I CLAIM:

1. A method of reducing noise in a signal, the method comprising the steps:

(1) supplying the input signal to an amplification unit;

(2) subjecting the input signal to an auxiliary noise reduction algorithm, to generate an auxiliary signal;

(3) using the auxiliary signal to determine a control input for the amplification unit; and

(4) controlling the amplification unit with the control signal, to generate an output signal with reduced noise.

2. A method as claimed in claim 1, wherein the input signal is subjected to a main noise reduction algorithm, to generate a modified input signal, which is supplied to the amplification unit.

3. A method as claimed in claim 2, wherein the main and auxiliary noise reduction algorithms are different.

704/233

4. A method of reducing noise in an input, audio signal containing speech, the method comprising the steps of:

(1) detecting the presence and absence of speech utterances;

(2) in the absence of speech, determining a noise magnitude spectral estimate;

(3) in the presence of speech comparing the magnitude spectrum of the audio signal to the noise magnitude spectral estimate;

(4) calculating an attenuation function from the magnitude spectrum of the audio signal and the noise magnitude spectral estimate; and

(5) modifying the input signal by the attenuation function, to generate an output signal with reduced noise.

5. A method as claimed in claim 4, wherein the square of the speech magnitude spectral estimate is determined by subtracting the square of the of the noise magnitude spectral estimate from the square of the magnitude spectrum of the input signal.

6. A method as claimed in claim 5, wherein the attenuation function is calculated in accordance with the following equation:

$$H(f) = \left[\frac{|X(f)|^2 - \beta\,|\hat{N}(f)|^2}{|X(f)|^2}\right]^{\alpha}$$

where $H(f)$ is the attenuation function, $|X(f)|$ is the magnitude spectrum of the input audio signal; $|\hat{N}(f)|$ is the noise magnitude spectral estimate, β is an oversubtraction factor and α is an attenuation rule, wherein α and β are selected to give a desired attenuation function.

7. A method as claimed in claim 6, wherein the oversubtraction factor β is varied as a function of the signal to noise ratio, with β being zero for high and low signal to noise ratios and with β being increased as the signal to noise ratio increases above zero to a maximum value at a predetermined signal to noise ratio and for higher signal to noise ratios β decreases to zero at a second predetermined signal to noise ratio greater than the first predetermined signal to noise ratio.

8. A method as claimed in claim 7, wherein the oversubtraction factor β is divided by a preemphasis function $P(f)$ to give a modified oversubtraction factor $\hat{\beta}(f)$, the preemphasis function being such as to reduce β at high frequencies, and thereby reduce attenuation at high frequencies.

9. A method as claimed in claim 6, 7 or 8 wherein the rate of change of

the attenuation function is controlled to prevent abrupt and rapid changes in the attenuation function.

10. A method as claimed in claim 6, wherein the attenuation function is calculated at successive time frames, and the attenuation function is calculated in accordance with the following equation:

$$G_n(f) = (1-\gamma)H(f) + \gamma G_{n-1}(f)$$

wherein $G_n(f)$ and $G_{n-1}(f)$ are the smoothed attenuation functions at the n'th and (n-1) 'th time frames, and γ is a forgetting factor.

11. A method as claimed in claim 10, wherein β is a function of perceptual distortion.

12. A method as claimed in claim 4 which includes remotely turning noise suppression on and off.

13. A method as claimed in claim 4 which includes automatically disabling noise reduction in the presence of very light noise or extremely adverse environments.

14. A method as claimed in claim 4 which includes detecting speech with a modified auto-correlation function.

15. A method as claimed in claim 14, wherein the auto-correlation function comprises:

(1) taking an input sample and separating it into short blocks and storing the blocks in correlation buffers;

(2) correlating the blocks with one another, to form partial correlations; and

(3) summing the partial correlations to obtain a final correlation.

16. A method a claimed in claim 15, wherein the method is carried out by digital signal processing and wherein the method includes using a Fast Fourier Transform to generate the partial correlations and includes detection of voiced speech directly in the frequency domain.

704/217

17. A method of determining the presence of speech in an audio signal, the method comprising taking a block of an input audio signal and performing an auto-correlation on that block to form a correlated signal; and checking the correlated signal for the presence of a periodic signal having a pitch corresponding to that for speech.

18. A method as claimed in claim 17, wherein the auto-correlation is performed on a first block taken from an audio signal, and a delayed block from the audio signal.

19. A method as claimed in claim 18, wherein each block is subdivided into a plurality of shorter sections and the correlation comprises a correlation between pairs of the shorter sections to form partial correlations, and subsequently summing the partial correlations to obtain the correlated signal.

20. A method as claimed in claim 19, wherein an input signal is stored as a plurality of samples in a pair of correlation buffers, and the auto-correlation is performed on the signals in the buffers to determine the partial correlations, which partial correlations are summed and stored.

21. An apparatus, for reducing noise in a signal, the apparatus including an input for a signal and an output for a noise reduced signal, the apparatus comprising:

(a) an auxiliary noise reduction means connected to the input for generating an auxiliary signal; and

(b) an amplification means connected to the input for receiving the original input signal and to the auxiliary noise reduction means, for receiving the auxiliary signal, the amplification means being controlled by the auxiliary signal to generate an output signal with reduced noise.

22. An apparatus as claimed in claim 21, wherein the auxiliary noise reduction means comprises:

(1) detection means connected to said input and providing a detection signal indicative of the presence of a desired audio signal;

(2) magnitude means for determining the magnitude spectrum of the input signal, with both the detection means and the magnitude means being connected to the input;

(3) spectral estimate means for generating a noise magnitude spectral estimate and being connected to the detection means and to the input of the apparatus; and

(4) noise filter calculation means connected to the spectral estimate means and the magnitude means, for receiving the noise magnitude spectral estimate and magnitude spectrum of the input signal to produce the auxiliary signal and having an output for the auxiliary signal connected to the amplification means.

23. An apparatus as claimed in claim 22, which includes a frequency transform means connected between said input and both of the magnitude means and the spectral estimate means for transforming the signal into the frequency domain to provide a transformed signal wherein the magnitude means determines the magnitude spectrum from the transformed signal, and wherein the spectral estimate means determines the noise spectral estimate from the transformed signal.

24. An apparatus as claimed in claim 23, wherein the noise filter calculation means determines the square of the speech magnitude spectral

estimate by subtracting the square of the noise magnitude spectral estimate from the square of the magnitude spectrum of the input signal and wherein the noise filter calculation means calculates the auxiliary signal as an attenuation function in accordance with the following equation:

$$H(f) = \left[\frac{|X(f)|^2 - \beta\,|\hat{N}(f)|^2}{|X(f)|^2} \right]^{\alpha}$$

where $H(f)$ is the attenuation function, $|X(f)|$ is the magnitude spectrum of the input audio signal; $|\hat{N}(f)|$ is the noise magnitude spectral estimate, β is an oversubtraction factor and α is an attenuation rule, wherein α and β are selected to give a desired attenuation function.

Add A18